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IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

Applicants:	Mirjana POPOVIC et al
Serial No.:	10/722,472
Filed:	November 28, 2003
For:	METHOD OF CAPTURING CONSTANT ECHO PATH INFORMATION IN A FULL DUPLEX SPEAKERPHONE
Group:	To Be Assigned
Examiner:	To Be Assigned

CLAIM FOR PRIORITY

Commissioner for Patents
P.O. Box 1450
Alexandria, VA 22313-1450

March 8, 2005

Sir:

Under the provisions of 35 U.S.C. §119 and 37 C.F.R. §l.55, Applicants hereby claim the right of priority based on:

British Patent Appln. No. 0227885.1, filed November 29, 2001.

A certified copy of the British Application is attached.

Respectfully submitted,

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Attachments
DES:dlh

The Patent Office

INVESTOR IN PEOPLE

The Patent Office
Concept House
Cardiff Road
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NP10 8QQ

I, the undersigned, being an officer duly authorised in accordance with Section 74(1) and (4) of the Deregulation & Contracting Out Act 1994, to sign and issue certificates on behalf of the Comptroller-General, hereby certify that annexed hereto is a true copy of the documents as originally filed in connection with the patent application identified therein.

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Dated 22 October 2003

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THE PATENT OFFICE
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02DEC02 E767470-1 D02917
P01/7700 0.00-0227885.1

Request for grant of a patent

The Patent Office
Cardiff Road
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South Wales NP10 8QQ

1. Your reference
5453001/CF

2. Patent Application Number
29 NOV 2001 0227885.1

3. Full name, address and postcode of the or of each applicant *(underline all surnames)*

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Patents ADP number *(if known)* 08508616001

If the applicant is a corporate body, give the country/state of its incorporation
Country: CANADA

4. Title of the invention
METHOD OF CAPTURING CONSTANT ECHO PATH INFORMATION IN A FULL DUPLEX SPEAKERPHONE USING DEFAULT COEFFICIENTS

5. Name of agent
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Patents ADP number 00001826001

6. Priority details

Country	Priority application number	Date of filing

0030Q2 E5T47O-1 D02917
P01/7700 0.00-0227852.1

Patents Form 1/77

7. If this application is divided or otherwise derived from an earlier UK application give details

Number of earlier application | Date of filing

8. Is a statement of inventorship and or right to grant of a patent required in support of this request?

YES

9. Enter the number of sheets for any of the following items you are filing with this form.

Continuation sheets of this form	
Description	10
Claim(s)	3
Abstract	1
Drawing(s)	3 + 3

10. If you are also filing any of the following, state how many against each item.

Priority documents	
Translations of priority documents	
Statement of inventorship and right to grant of a patent *(Patents form 7/77)*	1 + 2 copies
Request for preliminary examination and search *(Patents Form 9/77)*	1
Request for Substantive Examination *(Patents Form 10/77)*	
Any other documents *(please specify)*	

11. I/We request the grant of a patent on the basis of this application

Signature ______________________ Date 29 November 2002
BERESFORD & Co

12. Name and daytime telephone number of person to contact in the United Kingdom

FLEGG; Christopher Frederick

Tel: 020 7831 2290

Patents Form 7/77
Patents Act 1977
(Rule 15)

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Statement of inventorship and of right to grant of a patent

The Patent Office
Cardiff Road
Newport
South Wales NP10 8QQ

1. Your reference
5453001/CF

2. Patent Application Number
accompanying application reference 5453001
29 NOV 2001
0227885.1

3. Full name of the or each applicant
Mitel Knowledge Corporation

4. Title of the invention
METHOD OF CAPTURING CONSTANT ECHO PATH INFORMATION IN A FULL DUPLEX SPEAKERPHONE USING DEFAULT COEFFICIENTS

5. State how the applicant(s) derived the right from the inventor(s) to be granted a patent
by virtue of employment

6. How many, if any additional Patents Forms 7/77 are attached to this form?

7. I/We believe that the person(s) named over the page (and on any extra copies of this form) is/are the inventor(s) of the invention which the above patent application relates to.

Signature BERESFORD & Co   Date 29 November 2002

8. Name and daytime telephone number of person to contact in the United Kingdom
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Patents Form 7/77

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METHOD OF CAPTURING CONSTANT ECHO PATH INFORMATION IN A FULL DUPLEX SPEAKERPHONE USING DEFAULT COEFFICIENTS

Field of the Invention

The present invention relates in general to speakerphones and more particularly to a method of capturing constant echo path information in a full duplex handsfree (FDHD) speakerphone.

Background of the Invention

One of the most important performance indicators for full duplex speakerphones is convergence time (i.e. the time required by the echo cancellers within the speakerphone to reach an acceptable level of cancellation). The convergence time of the speakerphone depends both on internal Line Echo Canceller (LEC) and Acoustic Echo Canceller (AEC) convergence times. In order to converge quickly and properly, a speakerphone echo canceller requires a reference signal with correct stochastic properties. At the beginning of a call (Start-up), the reference signal is usually not sufficiently stochastic (e.g. the line signal typically comprises narrow band tones such as dial tone) or speech is not present, so that echo cancellation is unable to commence immediately. In such situations the speakerphone loop may remain unstable for a noticeable period of time. This can result in feedback or "howling" of the speakerphone during start-up, especially when the speaker volume is high.

In order to prevent such feedback, it is an objective of speakerphone design to ensure that the echo cancellers (LEC and AEC) converge rapidly to the correct echo path models at start-up. Otherwise, the speaker volumes must be reduced during start-up, which may be annoying to a user.

According to one prior art approach to reducing the problem of feedback during speakerphone start-up, howling detection has been used (see ITU-T Recommendation G.168) in combination with gain control. According to this approach, the speaker volume (or loop gain) is reduced when howling is detected. A

drawback of this approach is that the gain switching is often audible which may be annoying to the user.

Another prior art solution involves operating the speakerphone in a half duplex mode on start-up in order to prevent howling and echo from interfering with communication. The speakerphone remains in the half-duplex mode until the LEC adapts sufficiently to ensure echo cancellation. A drawback of this approach is that the speakerphone sometimes stays in the half-duplex mode for a long time, making communication between telephone parties difficult or impossible.

Yet another prior art solution involves forcing the speakerphone to start operation at a predetermined "acceptable" low volume level which guarantees stability in the audio loop, and then gradually increasing the volume as convergence of the echo canceller is achieved. A drawback of this approach is that the volume adjustment is often noticeable to the user.

Since the LEC models a network echo path where the first echo reflection of the near end hybrid is usually reasonably constant for each connection, and the AEC models an acoustic echo path where direct acoustic coupling or coupling through the plastic housing of the phone is always the same for a given phone, both the LEC and AEC may be loaded initially with previously captured and saved constant echo path models represented by default coefficients, and then continue to converge toward the complete echo channel models. This results in faster convergence time, and more stability as the main, strongest echo reflections will already be cancelled using the default coefficient models.

Thus, according to copending Patent Canadian Patent Application No. 2,291,428, a method is provided for improving the start-up convergence time of the LEC filter, thereby resulting in a total reduced convergence time for the speakerphone. This method is based on capturing the LEC coefficients once the LEC has converged, and saving them as the default coefficients for the next call. As a result, the echo-canceling algorithm does not have to wait for a suitable reference signal to commence convergence. At start-up, the echo canceller immediately begins canceling the line

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echo, based on the previously stored LEC coefficients, thereby assisting the AEC algorithm by eliminating residual line echo from the acoustic signal which the AEC algorithm is required to converge to, and initially making the speakerphone loop more stable. As indicated above, the same principal may also be applied to the AEC for direct acoustic coupling or coupling through the speakerphone housing plastic, which is always the same for a given phone. The default coefficients in this case represent the constant acoustic echo path from loudspeaker to microphone and may be reused for each new call. At start-up, the AEC immediately starts canceling the echo caused by direct acoustic coupling, while converging toward the complete acoustic echo path model that represents the combination of direct coupling and the specific room echo response.

The principle of saving default coefficients may also be applied to multiple loudspeaker-to-microphone echo paths for multiple-microphone directional systems, or even loudspeaker-to-beam echo paths for beamforming-based systems that perform echo cancellation on the output signal of a beamformer. In these cases, default coefficients can be reused from one instance to the next in each different direction (e.g. angular sectors).

In order for such systems to work properly, the coefficients must be saved at appropriate times. If they are saved at arbitrary instants (e.g. at the end of a call), then there is a risk that the full-duplex echo cancellation algorithm will not be in a well-converged state at the instant of saving the coefficients. For example, the echo cancellation algorithm may be in the process of adapting to an echo path change related to the user moving his/her hand towards the telephone to press a button for ending the call. Saving the default coefficients in this case and reusing them at a later stage (e.g. for the next call) may result in poor echo canceller performance until it re-converges to a set of "good" coefficients.

As indicated above, the system set forth in Canadian Patent Application No. 2,291,428 tracks the degree of convergence of the full-duplex algorithm, and saves the default coefficients each time the convergence reaches a predetermined level. In one embodiment, the amount of echo actually cancelled by the algorithm is measured, and

the coefficients are saved each time this amount increases by 3dB from the previous save. One problem with this method is that if the full-duplex algorithm is subjected to narrow-band signals (e.g. in-band tones that are not detected fast enough), then it may reach excellent levels of convergence with coefficients that are very different from the useful wide-band echo-path coefficients. In such situations the system may never reach as good a level of convergence again with a wider-band signal, such that proper coefficients are never captured. This may result in annoying echo bursts for the far-end user each time these coefficients are used (for instance, at the beginning of each subsequent call). Another problem is that if the telephone is moved to a different location on a desk, where the direct echo path is more difficult to adapt to, then it may never be able to capture coefficients corresponding to its new location. It may therefore constantly reuse coefficients that do not correspond to those characterizing the real echo path, resulting in mediocre echo cancellation until the algorithm has a chance to re-converge to the real echo path.

Summary of the Invention

According to the present invention, a method is provided for determining when to save coefficients so as to ensure that the system always captures coefficients that correspond to the best possible echo cancellation in its current condition, and to recover from scenarios where 'bad' default coefficients are captured. Thus, the saving of coefficients occurs at varying times depending on the amount of echo removed by the echo canceller. More particularly, the inventive method involves constantly monitoring the error signal to the echo canceller and comparing it with the error signal that would be obtained if default coefficients were to be used instead of the current coefficients. This ensures that the default coefficients are upgraded each time the current set of coefficients is better than the saved default coefficients.

Brief Description of the Drawings

A detailed description of the prior art and of a preferred embodiment of the invention is provided herein below with reference to the following drawings, in which:

Figure 1 is a block diagram of a prior art speakerphone echo canceller structure;

Figure 2 is a flow chart showing the steps of the echo cancellation method according to Applicant's own prior art; and

Figure 3 is a block diagram showing an adaptive filter structure for implementing a method of triggering capture of coefficients according to the present invention.

Detailed Description of Prior Art and Preferred Embodiment

As discussed briefly above, a speakerphone echo canceller comprises two adaptive filters which attempt to converge to two different echo models (acoustic and network echo) at the same time. As a result, speakerphones can easily become unstable, especially during start-up.

A traditional speakerphone echo canceller is shown in Figure1, wherein essential speakerphone components which are not related to echo cancellation have been omitted for clarity (e.g. double talk detector, non-linear processor, etc.) and are not addressed herein since they are not germane to the invention. The echo canceller attempts to model the transfer function of the echo path by means of an LEC filter and an AEC filter. The received signal (line or acoustic) is applied to the input of each filter (LEC and AEC) and to the associated echo path (network or acoustic) such that the estimated echo can be canceled by simply subtracting the signal which passes through each echo canceller from the received signal. If the transfer function of the model of the echo path is exactly the same as the transfer function of the echo path, the echo signal component is completely canceled (i.e. the error signal will be zero). The error signal is used for adaptation, so that the echo canceller converges to the correct transfer function, as discussed briefly above.

Typically, an algorithm such as the NLMS (Normalized-Least-Mean-Squared) algorithm is used to approximate the echo path (see "C261(UNIC) DSP Re-engineering and Performance Report" Mitel Semiconductor, Document No. C261AP13, Oct. 21, 1996).

From Figure 1 it will be appreciated that the residual echo after imperfect cancellation by the LEC will pass to the AEC reference signal. Since this residual echo is not correlated to the AEC received signal, this can cause the AEC filter to diverge. The extent to which AEC filter diverges depends on the level of the residual line echo. If the line echo is sufficiently canceled, its effect on the AEC behavior will be negligible.

Echo Return Loss Enhancement (ERLE) is an indicator of the amount of echo removed by an echo canceller. The ERLE is defined as:

$$ERLE(dB)=10\log_{10}[Power(ReceivedSignal)/Power(ErrorSignal)];$$

A generally acceptable LEC convergence time requires that the echo canceller achieve 27dB of ERLE in 0.5 sec (in ideal conditions).

Since the telephone is always connected to the same local loop (i.e. to the near-end Central Office (CO) or PBX), the impedance of the local loop remains the same for each call and consequently the near-end echoes remain fairly constant, from call to call. Accordingly, the local loop echo coefficients can be stored and re-used from call to call, thereby improving the start-up ERLE of the LEC. Furthermore, since the direct acoustic coupling through the plastic from loudspeaker to microphone is constant for given phone, the coefficients representing this part of the acoustic echo path can also be stored and re-used from call to call, thereby improving the start-up ERLE of the AEC.

Thus, with reference to the flowchart of Figure 2, which shows operation of the method set forth in Canadian Patent Application No. 2,291,428, after start-up of the echo canceller (Step 200), any previously stored default LEC coefficients are

loaded into the LEC. Although Canadian Patent Application No. 2,291,428 refers only to default coefficients being saved for the LEC, as indicated above the same principles apply to the AEC coefficients. Thus, the LEC (and/or AEC) begin(s) convergence using the well known NLMS algorithm (or other). On initial power-up of the speakerphone (i.e. prior to placing the first call), the initial coefficients are zero. Thus, the first call after power-up will always be a "training" call that results in capturing a suitable set of default coefficients for future calls. Next, at step 201,the "Call" proceeds. Signal levels of the LEC (and/or AEC) received signal and error signal are detected (step 203) and the ERLE is calculated using the formula set forth above (step 205). When a predetermined ERLE threshold level (Th) is reached (e.g. at least 24dB of echo is canceled), as calculated at step 207, and provided that the best LEC (and/or AEC) coefficients have not been previously saved during the call-in-progress (step 209), then the LEC (and/or AEC) coefficients of the (near) constant echo path are saved (step 211). Convergence of the LEC (and/or AEC) then proceeds as per usual and the call is completed (step 213). Once saved, the default coefficients are not recalculated again for the duration of the call (i.e. a YES decision at step 209). However, the LEC (and/or AEC) default coefficients will be calculated once per each call to ensure the best default set is captured for the next call.

At start-up of the next call, the previously stored LEC (and/or AEC) coefficients are retrieved and used as the default coefficient set for the LEC (and/or AEC) (step 200), instead of starting from zero.

The following pseudo code illustrates the principles of the above method in greater detail, wherein "EC" is used to indicate both the LEC and AEC:

```
Power-up: Default_coefficients = [000...0];

Start_Call:  EC_coefficients = Default_coefficients;
 Call:
                Execute EC algorithm;
                  Calculate power level of received signal ;
                  Calculate power level of error signal;
                  If (ERLE > Threshold) AND ( Best default set not saved)
                      Save near echo coefficients
               If Not(End of the Call) Go to Call;
```

If New Call Go to Start_Call;

Thus, each call subsequent to the initial power-up "training" call is provided with default coefficients that model the network and acoustic echo paths and guarantee small LEC and AEC error. This improves the training and tracking characteristic of the Full Duplex Handsfree Speakerphone (FDHF) and eliminates feedback during start-up. The best results are achieved when the training call uses a handset since there is no AEC-LEC loop instability and the LEC and AEC can therefore converge quickly.

According to the present invention, and in contrast with Applicant's prior method as set forth in Canadian Patent Application No. 2,291,428, instead of fixing the threshold ERLE at a value of 24 dB, the coefficients are captured and saved whenever there is an improvement in ERLE over a constantly increasing threshold value. In addition to this, the present invention constantly controls whether the saved set of the default coefficients is still valid. As shown in Figure 3, this is accomplished by continuously monitoring the error signal output from the subtractor (i.e. input signal – echo_current output from echo canceller) and comparing it to the error signal that would be obtained based on the default set of coefficients (i.e. input signal – echo_saved, depicted in Figure 3 using stippled lines). If the monitored error signal is less than that calculated using the default set of coefficients, then the current set of coefficients are saved as the new default set.

The following pseudo-code sets forth the method of the present invention in greater detail:

```
System Initialization: Default_coefficients=zeros;

Start_Call:  EC_coefficients = Default_coefficients;
 Call:
             Execute EC adaptation algorithm;
             Calculate error = input signal – echo_current;
              Calculate power level of received signal (Es);
              Calculate power level of error signal (Ee);
             Calculate ERLE as function of (Ee/Es);
             If (ERLE > Threshold)
Save:            Save Default_coefficients
                 Threshold= Threshold+ERLE_thr;  /* Increase the ERLE requirement by ERLE_thr dB*/
             End If

Check is executed HERE.
```

```
If Not(End of the Call) Go to Call;
If New Call Go to Start_Call;
```

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The foregoing is similar to the "Call" method set forth above with reference to Figure 2. The following "Check" algorithm ensures that the saved default coefficients are correct, according to the method of the present invention:

```
Check:
        Calculate error_saved = input signal – echo_saved;  /*Calculate error_saved using saved      */
                                                            /* default coefficients and rest of      */
                                                            /* coefficients in EC filter,            */
        Calculate power level of error_saved signal (Ee_saved);
        If (Ee < Ee_saved/ Error_thr)          /* Current coefficients are better than saved by Error_thr dB*/
        {
                Counter++;
                If (Counter == Time_threshold)
                {
                        Counter =0;
                        Save Default_coefficients;
                }
        }
        Else
        {
                Counter = Counter – DecThr; OR Counter=0;
        }
```

Thus, with each executions of the "Save" algorithm, the threshold is incremented until it reaches its maximum value for a given speakerphone. Then, the "Check" algorithm is used to correct or overwrite the default coefficients in the event that they have been incorrectly determined using the "Save" algorithm (e.g. due to narrow band training signal, phone being moved to a different location, etc.) Setting the ERLE_thr to be the same value as Error_thr , ensures that the "Save" algorithm results in saving the default coefficients while incrementing the threshold, and the "Check" algorithm re-saves the default coefficients only if the previously saved coefficients are no longer correct. In other words, the "Save" algorithm captures default coefficients whereas the "Check" algorithm verifies the saved coefficients.

According to the preferred embodiment, ERLE_thr = 6dB, Error_thr = 6dB, and Time_threshold = 2400 samples or 30ms.

Other embodiments and applications of the invention are possible. For example, this algorithm with some variations may also be implemented for the AEC filter to capture the acoustic feedback through the plastic, which will be constant for the

specific phone design. All such variations and modifications are believed to be within the sphere and scope of the invention as set forth in the claims appended hereto.

The present invention can be implemented by a computer program operating on a processor of an echo canceller. An aspect of the present invention thus provides a storage medium storing processor implementable instructions for controlling a processor to carry out the method as hereinabove described.

Further, the computer program can be obtained in electronic form for example by downloading the code over a network such as the internet. Thus in accordance with another aspect of the present invention there is provided an electrical signal carrying processor implementable instructions for controlling a processor to carry out the method as hereinbefore described.

We claim:

1. A method of updating of a set of default coefficients used for quick convergence of an echo canceller, wherein said echo canceller receives a reference signal and converges to an estimated echo signal of an input signal according to a current set of filter coefficients via feedback of a current error signal, said method including:

a) applying said default coefficients to said echo canceller for generating a further echo signal;

b) subtracting said further echo signal from said input signal to generate a further error signal; and

c) comparing said current error signal with said further error signal and in the event said current error signal exceeds said further error signal by a threshold amount then replacing said set of default coefficients by said current set of filter coefficients.

2. The method of claim 1, wherein said threshold amount is 6dB

3. The method of claim 1, wherein said set of default coefficients is replaced by said current set of filter coefficients only if said current error signal continuously exceeds said further error signal by said threshold amount of at least a predetermined period.

4. A method as claimed in claim 3 wherein the predetermined period is 30 ms.

5. An echo canceller for a speakerphone comprising means for updating of a set of default coefficients used for quick convergence of the echo canceller, wherein said echo canceller has means for receiving a reference signal and is operable to converge to an estimated echo signal of an input signal according to a current set of filter coefficients via feedback of a current error signal, the updating means comprising:

a) applying means for applying said default coefficients to said echo canceller for generating a further echo signal;

b) subtracting means for subtracting said further echo signal from said input signal to generate a further error signal; and

c) comparing means for comparing said current error signal with said further error signal and in the event said current error signal exceeds said further error signal by a threshold amount then replacing said set of default coefficients by said current set of filter coefficients..

6. The echo canceller of claim 5, wherein said threshold amount is 6dB.

7. The echo canceller of claim 5, wherein said set of

default coefficients is replaced by said current set of filter coefficients only if said current error signal continuously exceeds said further error signal by said threshold amount for at least a predetermined period.

8. The echo canceller of claim 7, wherein the predetermined period is 30ms.

9. A storage medium storing processor implementable instructions for controlling a processor to carry out the method of any one of claims 1 to 4.

10. An electrical signal carrying processor implementable instructions for controlling a processor to carry out the method of any one of claims 1 to 4.

ABSTRACT

METHOD OF CAPTURING CONSTANT ECHO PATH INFORMATION IN A FULL DUPLEX SPEAKERPHONE USING DEFAULT COEFFICIENTS

A method of determining when to save default coefficients in an echo canceller so as to ensure that the capture of coefficients that correspond to the best possible echo cancellation in a current condition. Coefficients are saved at varying times depending on the amount of echo removed by the echo canceller. More particularly, the present method involves constantly monitoring the error signal to the echo canceller and comparing it with the error signal that would be obtained if default coefficients were to be used instead of the current coefficients. This ensures that the default coefficients are upgraded each time the current set of coefficients is better than the saved default coefficients.

Figure 2 refers

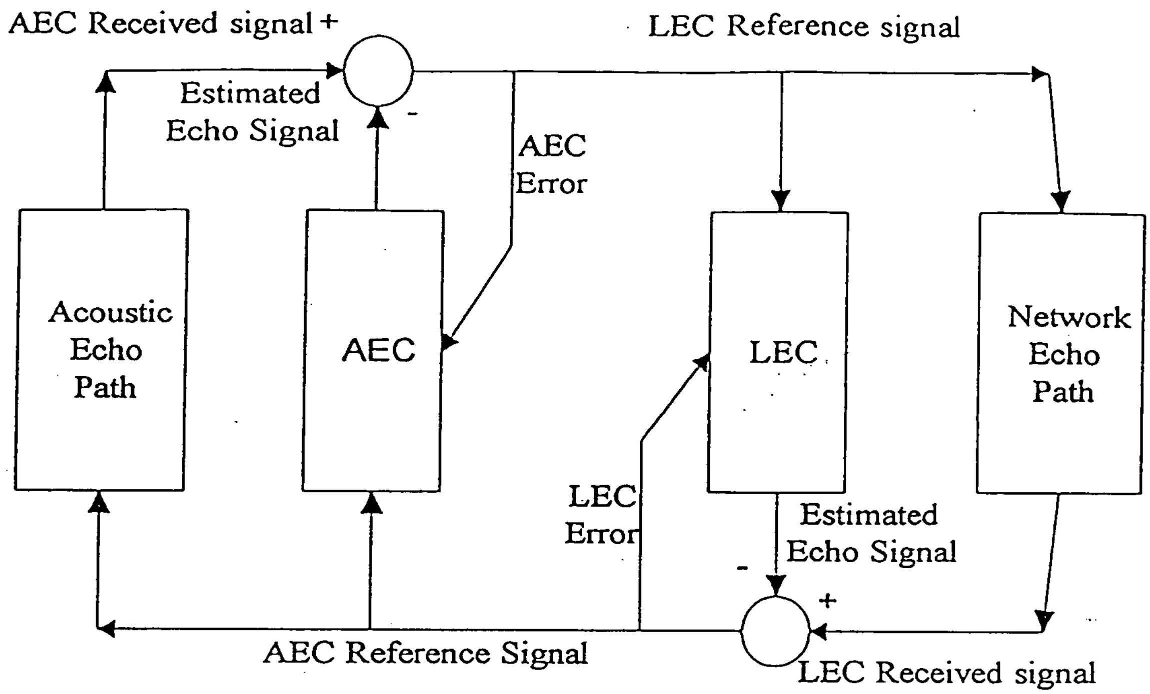
AEC Received signal +
LEC Reference signal
Estimated
Echo Signal
-
AEC
Error
Acoustic
Echo
Path
AEC
LEC
Network
Echo
Path
LEC
Error
Estimated
Echo Signal
-
+
AEC Reference Signal
LEC Received signal

Figure 1

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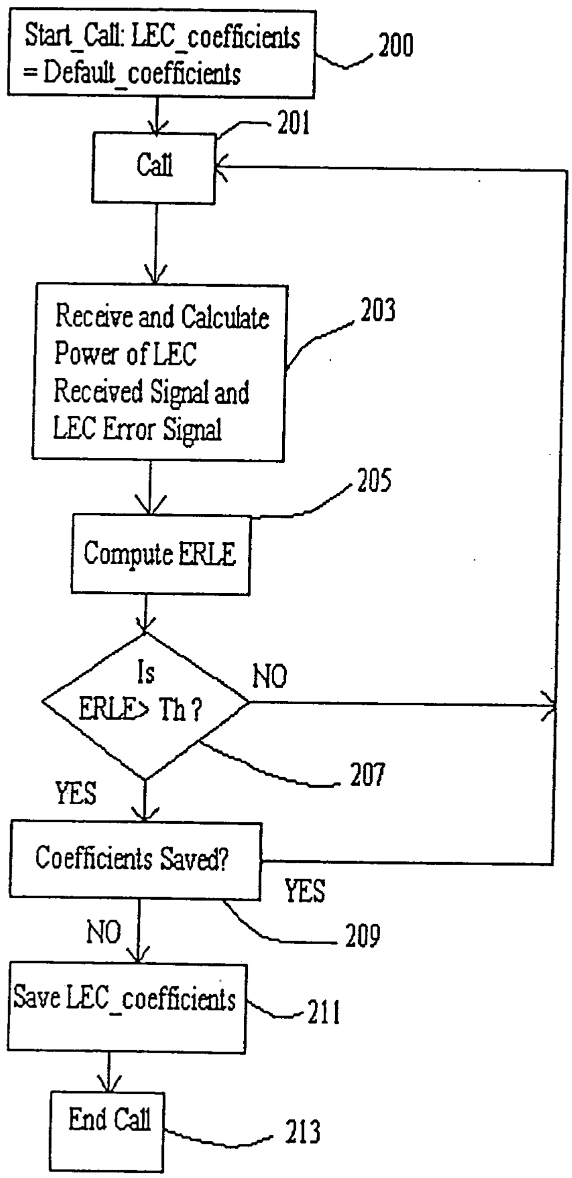
Start_Call: LEC_coefficients = Default_coefficients
200
Call
201
Receive and Calculate Power of LEC Received Signal and LEC Error Signal
203
Compute ERLE
205
Is ERLE> Th ?
NO
207
YES
Coefficients Saved?
YES
209
NO
Save LEC_coefficients
211
End Call
213

Figure 2

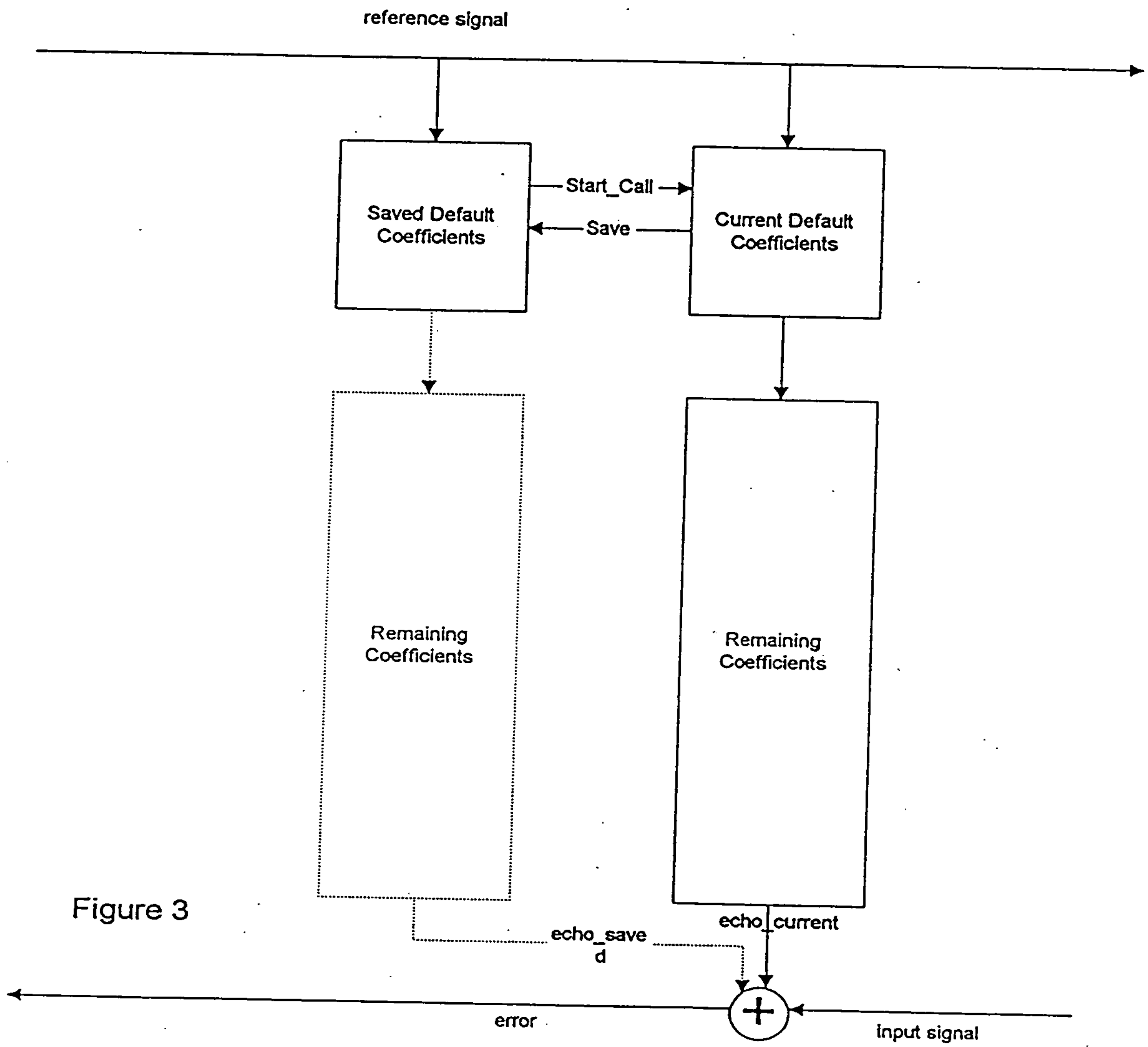
reference signal
Saved Default Coefficients
Start_Call
Save
Current Default Coefficients
Remaining Coefficients
Remaining Coefficients
echo_saved
echo_current
error
input signal

Figure 3

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USSN 10/722,472
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